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Analyzing Loudspeaker Locations for Sound Reinforcement Systems*

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By applying the concepts of critical distance, equivalent acoustic distance, needed acoustic gain, potential acoustic gain, and inverse square law attenuation to the analysis of the relative positioning of the talker, the listener, the microphone, and the loudspeaker in a total system which is to be Acousta-Voiced®, it is possible to accurately predict the actual performance of the proposed system. The acoustic gain needed for adequate performance, the acoustic gain actually available from the chosen configuration, and the required electrical power input to the loudspeaker are all easily calculated from basic formulas.

INTRODUCTION Every day our eyes and ears confirm that many engineers and technicians directly responsible for the design and installation of sound reinforcement systems experience difficulty in achieving efficient placement of loudspeakers in relation to the microphones used.

A successful sound reinforcement system should be able to provide the most remotely located listener with an acoustic signal having the same loudness and tonal balance found in close proximity to the live talker. Whether the ears of the remote listener are provided an acoustic signal equivalent to a location 2 ft or 10 ft in front of the talker depends on the ambient noise level and on the reverberation time. Experience has shown that listeners with normal hearing can clearly understand a talker, even in noisy and reverberant spaces, if they are within 2 to 10 ft of the talker. Acousta-Voicing[1] is demonstrably capable of providing control of tonal balance in a sound reinforcement system; therefore, this paper concentrates on achievement of sufficient acoustic gain and a usable ratio of direct to reverberant sound.

CRITICAL DISTANCE

A sound reinforcement system consists of a loop that includes the sound-system electronics, the electroacoustic transducers used, and the acoustic environment. The loop formed by these components will remain unconditionally stable until some frequency equals or exceeds unity gain when the phase angle is equal to $2\pi N$ (where $N = 0, 1, 2,$ etc.) [1]. From this fundamental limit, the acoustic gain of a sound reinforcement system can be calculated by ascertaining the acoustic level at which a signal from the loudspeaker reaches any open microphone in the system with an amplitude equal to that of the talker.

As the listening position is moved farther and farther from the loudspeaker, sound pressure level (SPL) delivered by the loudspeaker is attenuated inversely as the square of the distance:

$$20 \log_{10}(P_1/P_2) = \text{dB attenuation} \tag{1}$$

where P_1 = closest position and P_2 = farthest position.

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[1] Acousta-Voicing® is the process (patent pending) for the adjustment of the sound system's acoustic amplitude and phase response in contiguous critical band widths until equal acoustic feedback thresholds are achieved in each of the critical bands from 60 Hz to 16,000 Hz.

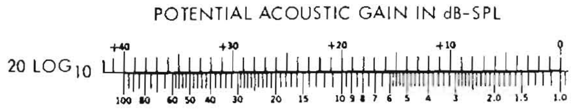

Fig. 6. Nomograph for potential acoustic gain according to Eq. (9).

$$20 \log_{10}[(28/1)(28/28)] = 29 \text{ dB}. \quad (13)$$

But in this case $\text{antilog}_{10}(23/20)$ no longer equals $[(28/1)(28/28)]$, and a new EAD should be calculated.

Whenever $\text{antilog}_{10}(\text{NAG}/20) \neq [(D_1/D_s)(D_0/D_2)]$, the EAD originally inserted in the NAG formula no longer holds true. A new EAD can be found as follows:

$$\text{EAD}/\text{antilog}_{10}[(\text{PAG}-\text{NAG})/20] = \text{new EAD}. \quad (14)$$

For a high-level system in the sample auditorium, the EAD is therefore found to be

$$8/\text{antilog}_{10}[(29-23)/20] = 4 \text{ ft}. \quad (15)$$

It would be naive to assume that the acoustic signal delivered to the remote listener with this EAD is exactly the same as what the listener would hear at 4 ft from the talker. While the SPL and tonal balance has been duplicated, the ratio of direct-to-reverberant sound has not been duplicated. Experience with more than 200 sound reinforcement systems where these calculations have been applied suggest that the $4D_c$ limit on D_2 is a realistic one.

DISTRIBUTION OF REINFORCED SOUND IN AUDITORIUMS

It should be realized that the benefits of maximum acoustic gain and controlled tonal balance hold true only for those audience areas that are uniformly covered by loudspeaker(s). In overhead distributed sound systems, this leads to assigning a 60° included angle to even the highest-quality coaxial loudspeakers, and then overlapping them 50% at the listener's ear level (see Fig. 7). In single-source systems, orientation of the horn, stuffing up to half the cells in a multicellular horn, and assigning differing power levels to various horns, are all useful methods in achieving uniform distribution. Where horns must be overlapped, it has been found that a 50% overlap is best as otherwise the fluctuations in pattern occurring at the higher frequencies can pose an insurmountable problem. In assigning differing powers to different horns, it should be remembered that D_c may well negate the desired effect for all but the horn covering the closest audience. Again, experience has indicated that a sound system remaining within ±2 or 3 dB in the 4000 Hz octave band anywhere in the audience area, where the system is excited with white noise, will benefit from Acousta-Voicing throughout the audience area.

ELECTRICAL POWER REQUIRED AT THE LOUDSPEAKER INPUT

The data developed thus far in analyzing potential loudspeaker locations in the hypothetical auditorium also serve to calculate the electrical input power necessary

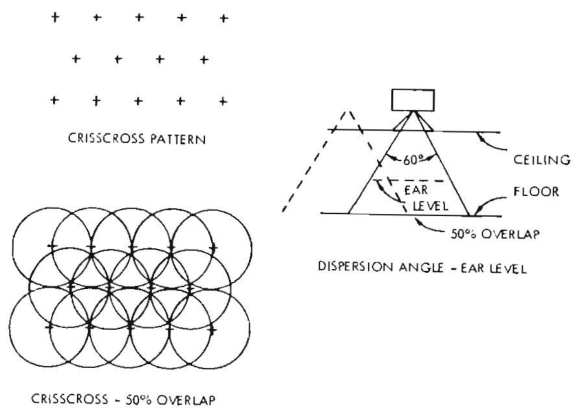

Fig. 7. Distributed loudspeaker data.

for each loudspeaker driver in the final system:

$$\text{antilog}_{10}\{[(SPL_T+10)+20 \log_{10}(D_1/4)-L_{EFF}]/10\}, \quad (16)$$

where (SPL_T+10) is the talker's SPL at 2 ft plus a 10 dB peaking factor, L_{EFF} is the efficiency rating of the loudspeaker measured with 1 W electrical input at 4 ft. For distances greater than D_c, substitute D_c for D_1.

The necessary electrical input power to the loudspeaker driver covering the remote listener in the sample auditorium can be calculated from Eq. (16) or read directly from the nomograph shown in Fig. 8. Calculating from the above equation

$$\text{antilog}_{10}[(80+10+17-99)/10] = 6.3 \text{ W}. \quad (17)$$

ROLE OF EFFICIENCY

The higher the efficiency and power-handling capability of the loudspeaker driver, the more dynamic will be the range available. For example, if the performer is not a talker but a rock-and-roll artist, situation will be represented by

$$\text{antilog}_{10}[(100+10+17-99)/10] = 630.9 \text{ W}. \quad (18)$$

The driver in this example has a program power rating of 40 W. The only practical solution is to find a driver with greater efficiency (while retaining the quality of the less efficient unit). Among high-quality drivers, an L_{EFF} of 112 dB is about maximum.

$$\text{antilog}_{10}[(100+10+17-112)/10] = 31.6 \text{ W}. \quad (19)$$

Just a little experience with the above formulae makes the sound engineer efficiency-minded.

TIME DELAY INTERFERENCE

Time delay interference is the major remaining variable that must be given consideration before a tentative

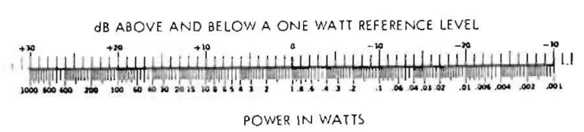

Fig. 8. Conversion of dB to power in W.

Within enclosed spaces, a critical distance D_c is rapidly reached where the transition from a directional sound field to a diffuse sound field occurs. D_c is defined as that distance from a sound source where the SPL of the direct sound and of the reflected sound are equal; hence for distances less than D_c inverse square law operates, and for distances greater than D_c the SPL tends to remain constant with increasing distance.

It should be carefully considered that while the measured SPL beyond the critical distance tends to remain constant with increasing distance, the direct sound continues to follow inverse square law attenuation. Good engineering practice for sound reinforcement systems is to insure that the direct sound does not drop more than 12 dB below the reflected energy.

D_c for a particular acoustic environment is found by:

$$0.14\sqrt{QS\bar{a}} = D_c \qquad (2)$$

where 0.14 is a constant, $\bar{a}$ is the average absorption coefficient of the room surfaces, and S is the total surface area of the space (walls, floor and ceiling) and Q = the directivity factor of the sound source. To illustrate this calculation, assume an enclosed space of 750,000 ft^3 with an average absorption coefficient = $\bar{a}$ 0.12 and a total surface area S = 55,000 ft^2:

$$0.14\sqrt{6 \times 0.12 \times 55000} = 28\ D_c \qquad (3)$$

Figure 1 shows these relationships for variations of $\bar{a}$ and S where the Q = 6 (typical of multicellular and sectoral horns commonly used in high quality sound reinforcement systems).

The D_c of the auditorium also determines the maximum distance from the loudspeaker to a remote listener (D_2). Since inverse square law attenuation operates for the total signal up to D_c and only for the direct sound beyond D_c, the direct sound will fall 12 dB below the reflected sound at a distance equal to $4D_c$. In the auditorium of our example, no D_2 distance greater than 112 ft (4 × 28 ft) should therefore be considered.

Figures 2 and 3 detail and label the pertinent relationships between the talker, the listener, the microphone, and the loudspeaker. Referring to the 750,000 ft^3 auditorium for which a D_c of 28 ft was obtained, D_o is found to be 100 ft. Where D_o is greater than D_c substitute D_c for D_o, except when measuring time delay (TD).

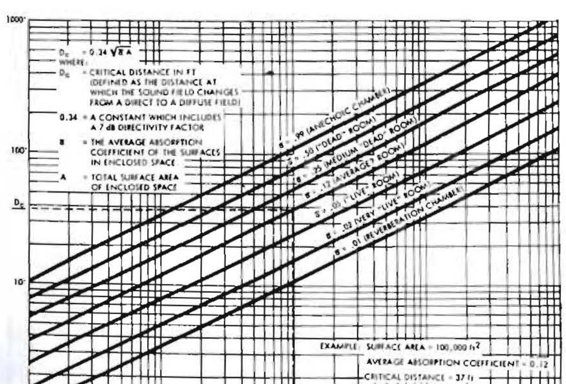

Fig. 1. Relationships of critical distance factors.

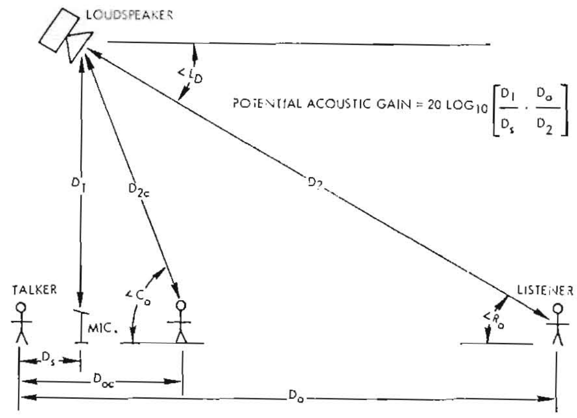

Fig. 2. Physical measurements used in performance calculations for single-source sound system.

EQUIVALENT ACOUSTIC DISTANCE

An equivalent acoustic distance (EAD) can be selected that provides excellent hearing conditions without a sound reinforcement system. To determine EAD, assume a higher than desired ambient noise level of NC45 and a 500 Hz, ⅓ octave band reverberation time of 3 sec. Because of these conditions, it is found that a distance of 8 ft from the talker is the maximum for completely relaxed listening. The goal, therefore, should be never to let the SPL at the remote listener drop below the SPL generated by the talker (without a sound system) at a distance of 8 ft. This distance is the EAD. The D_o loss in dB for an EAD of 8 ft can be calculated as follows:

$$20 \log_{10}(\mathrm{EAD}/D_0) = \text{loss in dB}; \qquad (4)$$

therefore,

$$20 \log_{10}(8/28) = -11 \text{ db}. \qquad (5)$$

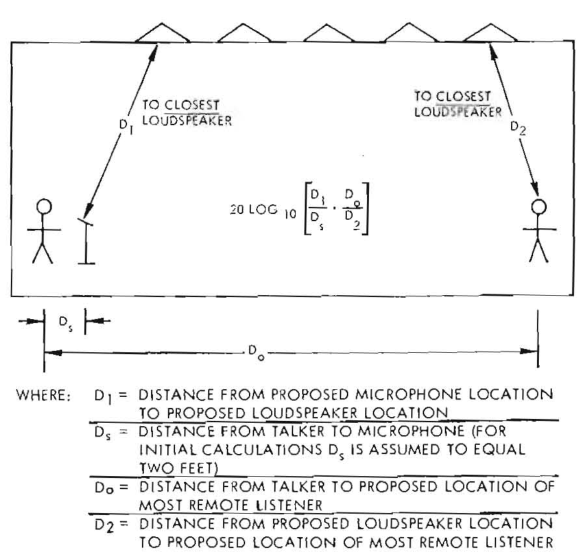

Fig. 3. Physical measurements used in performance calculations for distributed sound systems.

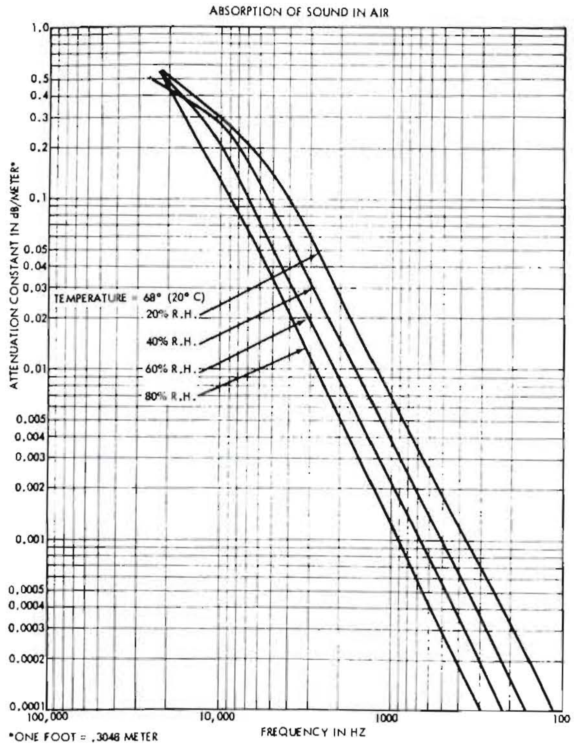

Fig. 4. Attenuation with respect to frequency due to humidity and air absorption.

The attenuation rate for frequencies greater than 1000 Hz, however, exceeds that predicted by inverse square law. This is caused by molecular air absorption and the effects of humidity. While this factor does not appreciably affect acoustic gain calculations, it does lead to a different tonal balance at a distance from the loudspeaker compared to a close position. If the amplitude response of the sound system is made uniform at 100 ft, then a listener at 40 ft will hear the high frequencies boosted. If the response at the listener closest to the loudspeaker is made uniform (the desired case), then those seats farther removed will show a gradually sloping response from 1000 Hz to 10,000 Hz. Fig. 4 illustrates the degree of additional attenuation to be expected at the higher frequencies [2].

Simultaneously Open Microphones

When more than one microphone is brought up on the mixer, the potential acoustic gain (PAG) is reduced. Each time the number of open microphones (NOM) is doubled, 3 dB of PAG is lost. This is shown in Fig. 5 and defined by:

$$10 \log_{10}(1/\mathrm{NOM}) = \text{loss of PAG in dB}. \quad (6)$$

Fig. 5. Effect of the number of open microphones on gain.

Note that all microphones are assumed to be adjusted the same number of dB below the feedback threshold and to be in the reverberant field of the loudspeaker.

NEEDED ACOUSTIC GAIN

There is now enough data on hand from this hypothetical auditorium to calculate the NAG:

$$[20 \log_{10}(D_0/\mathrm{EAD})] + (10 \log_{10}\mathrm{NOM}) + 6 = \mathrm{NAG}. \quad (7)$$

For any distance greater than D_c, substitute D_c. The $+6$ represents the gain reduction that will ultimately be required to stabilize the system [3]. This constant is included because all the calculations are based on unity gain, and the sound system actually cannot be operated at that level. Substituting the values of the example auditorium in Eq. 7, NAG is derived as

$$[20 \log_{10}(28/8)] + (10 \log_{10} 2) + 6 = 11\ \mathrm{dB} + 3\ \mathrm{dB} + 6\ \mathrm{dB} = 20\ \mathrm{dB}. \quad (8)$$

The basic formula for the potential acoustic gain (PAG) is:

$$20 \log_{10}[(D_1/D_s)(D_0/D_2)] = \mathrm{PAG}. \quad (9)$$

Since the NAG is now determined,

$$\mathrm{antilog}_{10}(\mathrm{NAG}/20) = (D_1/D_s)(D_0/D_2). \quad (10)$$

To place the sound system microphone in the reverberant field of the loudspeakers and to ensure the maximum benefit from their separation, it can be further stated that in the example auditorium D_1 should be not less than 28 ft.

The distance D_0 of 100 ft is already known; to determine the distance, D_s the intended use of the system must be examined. For instance, assume a moderator and three panelists on the stage. The moderator wears a lavalier and hence presents no problem, but the panel member farthest from the microphone on the table is 6 ft from the microphone. This is the distance D_s. If x is substituted for D_2, then

$$\mathrm{antilog}_{10}(20/20) = [(28/6)(28/x)]; \quad x \text{ (or } D_2) = 13 \text{ ft}. \quad (11)$$

For any distance greater than D_c, substitute D_c. This distance quickly reveals that an overhead distributed sound system is the only possible solution to the parameters presented.

For an alternative, assume the ceiling is too high and the realism of a single source system is desired. To analyze this idea, recalculate the NAG for four microphones instead of two, so that each of the three panelists and the moderator can have their own microphone. As can be seen from Fig. 5, an increase from 2 to 4 open microphones will add 3 dB more to the NAG figure. The new NAG therefore becomes 23 dB; if x is substituted for D_s, maximum D_s can be derived by calculation:

$$\mathrm{antilog}_{10}(23/20) = [(28/x)(28/28)]; \quad x = 2 \text{ ft} = \text{maximum } D_s. \quad (12)$$

This distance suggests lavalier microphones as an ideal solution. This would decrease D_s to 1 ft and the PAG (see Fig. 6) would then become

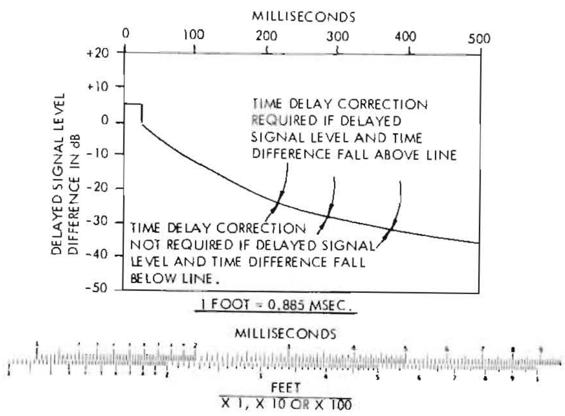

Fig. 9. Time delay correction curve.

loudspeaker location can be accepted as satisfactory. The single-source system normally avoids time delay problems, thanks to the nearly equal distances between the loudspeaker and all listeners.

Time delay must be inspected in the overhead distributed system or in the combined single-source overhead distributed system, and two parameters must be obtained and reconciled: arrival time difference, and difference in SPL between the delayed signal and the signal from the nearby loudspeaker. The SPL difference is the PAG at the listener position of interest. The TD is

$$.885\, D_0 - .885\, D_2 = \text{TD}. \tag{20}$$

In the auditorium of the example,

$$88.5 \text{ msec} - 88.5 \text{ msec} = 0, \tag{21}$$

where $D_0 = 100$ ft, $D_2 = 100$ ft, and PAG = 29 dB. Note that in the calculation of TD, D_r should *not* be substituted for distances.

Figure 9 shows the range of levels and delays; those below the curved line do not require a time delay mechanism and those above the curved line do require such a mechanism.

CONCLUSIONS

It has been shown it is possible to develop, from either drawings or a site survey, meaningful acoustic predictions of NAG, PAG and EAD. Sufficient data is generated from these calculations to allow further calculation of variations in microphone, listener, talker, and loudspeaker spacing, and to obtain the required electrical input power to the loudspeaker.

Finally, sound distribution control, time delay *vs* level effects, and the results of multiple microphone usage have been discussed. It is hoped that these simplifications of acoustic equations into a form believed accurate, within good engineering practices for sound reinforcement systems, will provide useful insights into the design of sound reinforcement systems capable of being successfully Acousta-Voiced.

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THE AUTHOR

Donald B. Davis was born in 1928 in Oak Park, Illinois and attended Purdue University. He has been with Altec Lansing, a division of LTV Ling Altec, in Anaheim, California since 1959 where he is presently manager of the company's Acousta-Voicing® operation.

Mr. Davis is a member of the Acoustical Society of America, Society of Motion Picture and Television Engineers, and the Audio Engineering Society. He recently served as AES Western Vice-President and Convention Chairman and is currently a member of its Board of Governors. Mr. Davis is the author of 2 books on audio engineering and numerous technical articles.